DAG 2000(24/32FXS) User Manual V1.0

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1. Equipment Introduction

1.1 Introduction

The DAG 2000(24/32FXS) is a full feature voice and fax-over IP device that offers a high-level of integration including NAT, 24/32 ports FXS telephone gateway and has market-leading sound quality, rich functionalities, and a compact and lightweight design. The DAG 2000(24/32FXS) fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market. Moreover, it supports comprehensive voice codec including G.723.1, G.729AB and PCMA/U. A typical network diagram shows the function of DAG 2000(24/32FXS) as below.

Figure 1-1-1 Application topology

1.2 Equipment Structure

1.2.1 Rear View

Figure 1-2-1 DAG 2000(32FXS) Rear View

Table 1-2-1 Description of DAG 2000(32FXS) Rear View

Interface	**Description**
PWR	Connecting to power adapter, 110~240VAC,50~60HZ,0.5A
Off/On	Power switch
Ground	Grounding

1.2.2 Front View

Figure 1-2-2 DAG 2000(32FXS) Front View

Table 1-2-2 Description of DAG 2000(32FXS) Front View

LED	**Color**	**Name**	**Status**	**Description**
POWER	Green	Power status indicator	Off	Power is off
			On	Power is on
RUN	Green	Register indicator	Fast blinking	Register
			Slow blinking	Unregister
WAN	Yellow	WAN status indicator	Off	Failed
			On	Normal
FXS	Green	Indicate status of the respective FXS ports	Off	Available
			On	Busy
RST	Restart equipment			
Console	Connected the PC with a serial line and used to configure equipment			

1.3 Connection to DAG 2000(24/32FXS)

The DAG 2000(24/32FXS) is easy to configure using the embedded GUI page and the following three steps.

Three easy steps to connect DAG 2000(24/32FXS)

1. Connecting a standard touch-tone analog telephone (or fax machine) to FXS port
2. Inserting the Ethernet cable into the WAN port of device and connecting the other end of the Ethernet cable to an uplink port (a router, PC or a modem, etc.)
3. Inserting the power adapter into the device and connecting it to a wall outlet

1.4 Functions and Features

1.4.1 Protocol standard supported

- Standard SIP and MGPC(option) protocol
- Simple Traversal of UDP over NATs (STUN)
- IP Transport: RTP/RTCP
- Hypertext Transfer Protocol (HTTP)
- Domain Name System (DNS)
- SNMP
- ITU-T G.723.1、G.729AB and PCMA/U

1.4.2 System function

- PLC,VAD and CNG
- DTMF mode: RFC 2833, SIP INFO and INBAND
- T.38/ Pass-Through FAX over IP
- Silence Suppression
- HTTP/Telnet configuration
- Firmware upgrade by Http
- Caller ID, Call waiting, Call transfer and DND
- Three-way conference

1.4.3 General hardware specification

- Power supply: Input 110~240 VAC/50~60HZ, 0.5A
- Temperature: 0~40℃(operational),-20~70℃(storage)
- Humidity: 10%~90%, no condensation
- Max power consumption:56W
- Dimension(mm): 440*250*44
- Net weight: 3.5kg

2. Basic Operations

2.1 Phone Call

2.1.1 Phone or Extension Numbers

1. Dial the number directly and wait for 3 seconds (Default "*No dial timeout*");
2. Dial the number directly and press #.

Examples:

1. Dial an extension directly on the same proxy, (e.g. 8080), and then press the # or wait for 3 seconds.
2. Dial an outside number (e.g. (626) 666-8080), first enter the prefix number (usually 1+ or international code) followed by the phone number. Press # or wait for 3 seconds. Check with your VoIP service provider for further details on prefix numbers.

2.1.2 Direct IP Calls

Direct IP calling allows two parties, that is, a FXS Port with an analog phone and another VoIP Device, to talk to each other in an ad hoc fashion without a SIP proxy.

Elements necessary to completing a Direct IP Call:

1. Both DAG serial (24/32FXS) and other VoIP Device, have public IP addresses;
2. Both DAG serial (24/32FXS) and other VoIP Device are on the same LAN using private IP addresses;
3. Both DAG serial (24/32FXS) and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

Processing:

1. Pick up the analog phone then dial "***47**"
2. Enter the target IP address.

Note: No dial tone will be played between step 1 and 2.

Examples:

If the target IP address is 192.168.0.160, the dialing convention is ***47,** then **192*168*0*160.** Followed by pressing the "#" key or wait 3 seconds. Destination port is 5060.

NOTE:

Customers cannot make direct IP calls between FXS0 to FXS1 since they are using same IP. It only supports the default destination port 5060. "*Disable direct IP-IP calling*" must be set to "No" in "System configuration" of web configuration page. "*Call features enable*" must be set to "Yes" in web configuration page.

2.2 Call Hold

Place a call on hold by pressing the "flash" button on the analog phone (if the phone has that button).Press the "flash" button again to release the previously held Caller and resume conversation. If no "flash" button is available, use "hook flash" (toggle on-off hook quickly). You may drop a call using hook flash.

2.3 Call Waiting

Call waiting tone (3 short beeps) indicates an incoming call, if the call waiting feature is enabled. Toggle between incoming call and current call by pressing the "flash" button. First call is placed on hold. Press the "flash" button to toggle between two active calls.

2.4 Call Transfer

2.4.1 Blind Transfer

Assume that call Caller A and B are in conversation. A wants to Blind Transfer B to C: 1. Caller A presses **FLASH** on the analog phone to hear the dial tone. 2. Caller dials ***87** then dials caller C's number, and then # (or wait for 4 seconds) 3. Caller A will hear the confirm tone. Then, A can hang up.

NOTE: "*Call features enable*" must be set to "Yes" in web configuration page. Caller A can place a call on hold and wait for one of three situations:

1. A quick confirmation tone (similar to call waiting tone) followed by a dial-tone. This indicates the transfer is successful. At this point, Caller A can either hand up or make another call.
2. A quick busy tone followed by a restored call. This means the transferee has received a 4xx response for the INVITE and we will try to recover the call. The busy tone is just to indicate to the transferor that the transfer has failed.
3. Continuous busy tone. The phone has timed out.

2.4.2 Attended Transfer

Assume that Caller A and B are in conversation. Caller A wants to *Attend Transfer* B to C:

1. Caller A presses **FLASH** on the analog phone for dial tone.
2. Caller A then dials Caller C's number followed by # (or wait for 3 seconds).
3. If Callee C answered the call and allowed transfer, then Caller A pressed FLASH again and hang up. Callee B and C are in conversation. Transfer is completed.
4. If Caller C does not answer the call or not allowed to transfer, then caller A can press "flash" to resume call with Caller B.

2.5 Call Features

DAG 2000(24/32FXS) supports all the traditional and advanced telephony features.

Table 2-5-1 DAG serial Call Feature Definition

*47	Direct IP Calling. Dial "*47" + "IP address". No dial tone is played in the middle.
*50	Disable Call Waiting(for all subsequent calls)
*51	Enable Call Waiting(for all subsequent calls)
*72	Unconditional Call Forward: Dial "*72" and then the forwarding number followed by "#". Wait for dial tone and hang up. (dial tone indicates successful forward)
*73	Cancel Unconditional Call Forward: To cancel "Unconditional Call Forward", dial "*73", wait for dial tone, and then hang up.
*78	Enable Do Not Disturb(DND): When enabled all incoming calls are rejected
*79	Disable Do Not Disturb(DND): When disabled, incoming calls are accepted.
*87	Blind Transfer
*90	Busy Call Forward: Dial "*90" and then forwarding number followed by "#".Wait for dial tone then hang up.
*91	Cancel Busy Call Forward. To cancel "Busy Call Forward", dial "*91",wait for dial tone, and then hang up.
*92	No Answer Call Forward. Dial "*92" and then the forwarding number followed by "#". Wait for dial tone then hang up.
*93	Cancel No Answer Call Forward. Dial "*93", wait for dial tone, and then hang up.
*99	Cancel all call forward
*114#	Report phone No
*158#	Report IP Address
*111#	Reset
Flash/Hook	Toggles between active call and incoming call(call waiting tone). If not in conversation, flash/hook will switch to a new channel for a new call.

2.6 Sending and Receiving Fax

DAG 2000(24/32FXS) supports fax in two modes: 1) T.38 (Fax over IP) and 2) fax pass through. T.38 is the preferred method because it is more reliable and works well in most network conditions. If the service provider supports T.38, please use this method by selecting T.38 as fax mode (default). If the service provider does not support T.38, pass-through mode may be used. If you have problems with sending or receiving Fax, toggle the Fax Tone Detection Mode setting.

3. Configuration Guide

3.1 IP Address of the device configuration

Modify the device IP in two ways: (1) Voice over the telephone to configure; (2) Through the serial port to configure.

3.1.1 Configuration with telephone

1. Dynamic IP address by DHCP：
   Offhook;
   Input “*150*2#”;
   Onhook;
   If the equipment hint success, after 10 seconds, and restart the equipment.
2. Static IP address：
   Offhook;
   Input “*150*1#”;
   Onhook;
   Then
   （1） Configure IP address：
   Offhook; input “*152*172*16*0*100# ”; onhook.
   （2） Configure netmask：
   Offhook; input “*153*255*255*0*0# ”; onhook.
   （3） Configure gateway IP address (next hop)：
   Offhook; input “*156*172*16*0*1# ”; onhook.
   （4） if success, after 10 seconds, restart the equipment ;
3. Query the IP address of device : Offhook, input “*158#”
4. If the device uses PPPoE method to get IP address，it need to configure by web browser.

NOTE: the telephone will play voice hint “Setting successfully” if the setting step is correct.

3.1.2 Configuration with serial port

1. Device console port connected to the PC via the serial line.
2. Open super terminal and set baud rate to 115200bps. Then login the device via command line.
3. Customers can modify the device IP on the command line.

3.2 Entrance DAG 2000(24/32FXS) By Web Browser

The device has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allow users to configure the device through a web browser such as Microsoft IE .

3.2.1 Login

The device GUI configuration can be accessed via WAN port:

1. Offhook, input ***158#** and get an IP address. The default gateway IP address: 192.168.11.1.
2. Directly connect a computer to WAN port with cable.
3. Open a command window on the computer.
4. Type in "ipconfig /release", the IP address becomes 0.
5. Type in "ipconfig /renew", the computer gets an IP address in 192.168.11.x segment by default
6. Open a web browser, input the gateway IP address 172.30.66.20. Customers will see the login page of the device as figure3-2-1.
7. Entry the default username and password "admin" and click OK, then customers can enter the configuration page as figure3-2-2.

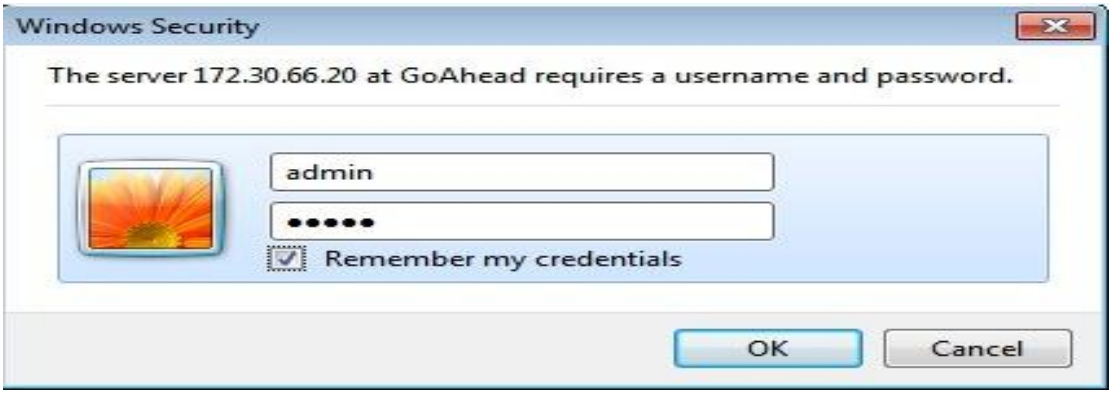

Figure 3-2-1 Login

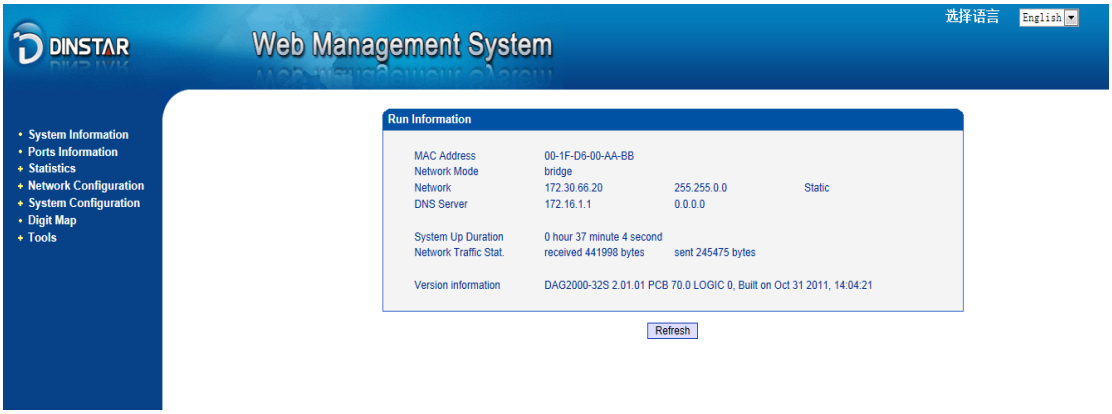

Figure 3-2-2 Configuration Page

According to their own situation, users can select Chinese or English interface in the upper right corner of the page. Go through navigation tree, user can check, modify, and set the device configuration on the right of configuration interface.

3.3 System Information

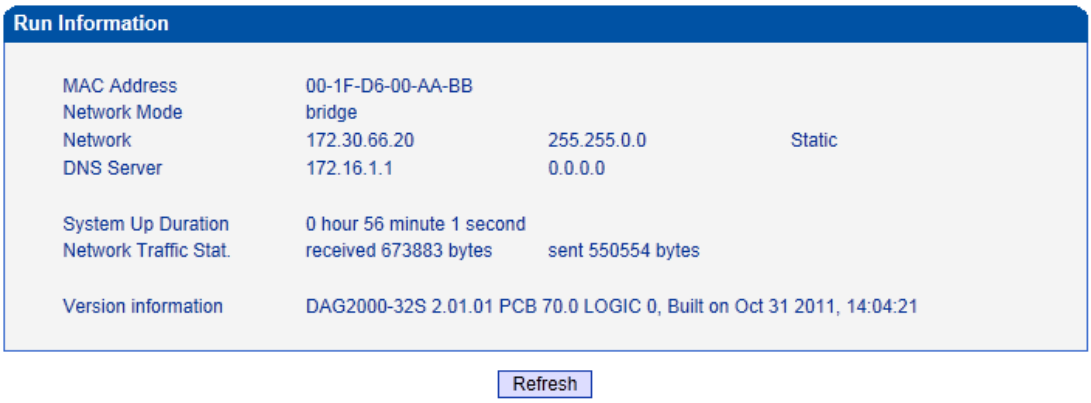
Run Information
MAC Address 00-1F-D6-00-AA-BB
Network Mode bridge
Network 172.30.66.20 255.255.0.0 Static
DNS Server 172.16.1.1 0.0.0.0
System Up Duration 0 hour 56 minute 1 second
Network Traffic Stat. received 673883 bytes sent 550554 bytes
Version information DAG2000-32S 2.01.01 PCB 70.0 LOGIC 0, Built on Oct 31 2011, 14:04:21
Refresh

Figure 3-3-1 System Information

Table 3-3-1 Description of System Information

MAC address	WAN port hardware address
Network Mode	Displaying the current network mode, route mode or bridge mode.
Network	Displaying WAN port IP address, subnet mask, default gateway and the way of obtain IP address.
DNS Server	Displaying IP address and default gateway of primary DNS server
System Up Duration	Time elapsed from device power on to now.
Network Traffic Statics	Total bytes of message received and sent by network port
Version Information	Includes: product mode, software version, hardware version and built time etc.

3.4 Ports Information

Ports Information

Port No.	Type	Registered	Number
0	FXS	Yes	201
1	FXS	Yes	202
2	FXS	No	
3	FXS	No	
4	FXS	No	
5	FXS	No	
6	FXS	No	
7	FXS	No	
8	FXS	No	
9	FXS	No	
10	FXS	No	
11	FXS	No	
12	FXS	No	
13	FXS	No	
14	FXS	No	
15	FXS	No	
16	FXS	No	
17	FXS	No	
18	FXS	No	
19	FXS	No	
20	FXS	No	
21	FXS	No	
22	FXS	No	
23	FXS	No	
24	FXS	No	
25	FXS	No	
26	FXS	No	
27	FXS	No	
28	FXS	No	
29	FXS	No	
30	FXS	No	
31	FXS	No	

Refresh

Figure 3-4-1 Ports Information

Ports Information display some port information, include: Port No, Type, Registered, Number. Intuitive status display makes ports management more convenient.

3.5 Statistics

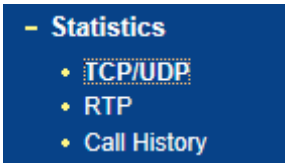

Figure 3-5-1 Statistics

Statistics Information includes: TCP/UDP, RTP and Call History statistics.

3.5.1TCP/UDP

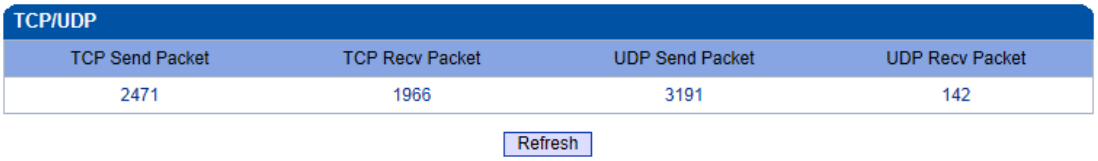

Figure 3-5-2 TCP/UDP

TCP/UDP statistics is used to statistic the number of sending and receiving TCP/UDP packages.

3.5.2 RTP

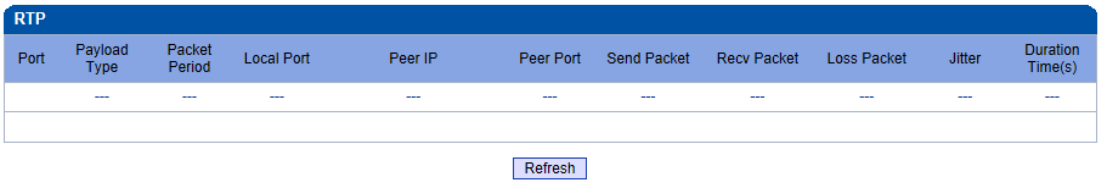

Figure 3-5-3 RTP

Table 3-5-1 Description of RTP

Port	FXS port ID
Payload Type	Codec mode(eg:G.723.1, G.729, PCMA/U)
Packet Period	Time of packaging
Local Port	Local port of transmitting RTP packages
Peer IP	End of equipment IP address
Peer Port	Peer port of receiving RTP packages
Send Packet	Total of sending RTP packages
Recv Packet	Total of receiving RTP packages
Loss Packet	Total of losing RTP packages
Jitter	Length of delay jitter
Duration Time(s)	Both ends of the call time

3.5.3 Call History

This option records call history information of every port.

Call History

Port	Incoming Received	Incoming Connected	Incoming Answered	Incoming Failed	Outgoing Attempted	Outgoing Connected	Outgoing Answered	Outgoing Failed
0	6	6	1	5	2	1	1	1
1	1	1	1	0	6	6	1	5
2	0	0	0	0	0	0	0	0
3	0	0	0	0	0	0	0	0
4	0	0	0	0	0	0	0	0
5	0	0	0	0	0	0	0	0
6	0	0	0	0	0	0	0	0
7	0	0	0	0	0	0	0	0
8	0	0	0	0	0	0	0	0
9	0	0	0	0	0	0	0	0
10	0	0	0	0	0	0	0	0
11	0	0	0	0	0	0	0	0
12	0	0	0	0	0	0	0	0
13	0	0	0	0	0	0	0	0
14	0	0	0	0	0	0	0	0
15	0	0	0	0	0	0	0	0
16	0	0	0	0	0	0	0	0
17	0	0	0	0	0	0	0	0
18	0	0	0	0	0	0	0	0
19	0	0	0	0	0	0	0	0
20	0	0	0	0	0	0	0	0
21	0	0	0	0	0	0	0	0
22	0	0	0	0	0	0	0	0
23	0	0	0	0	0	0	0	0
24	0	0	0	0	0	0	0	0
25	0	0	0	0	0	0	0	0
26	0	0	0	0	0	0	0	0
27	0	0	0	0	0	0	0	0
28	0	0	0	0	0	0	0	0
29	0	0	0	0	0	0	0	0
30	0	0	0	0	0	0	0	0
31	0	0	0	0	0	0	0	0

Refresh

Figure 3-5-4 Call History

Call history statistics show number 0~31 port inbound and outbound call statistics.

3.6 Network Configuration

3.6.1 Local Network

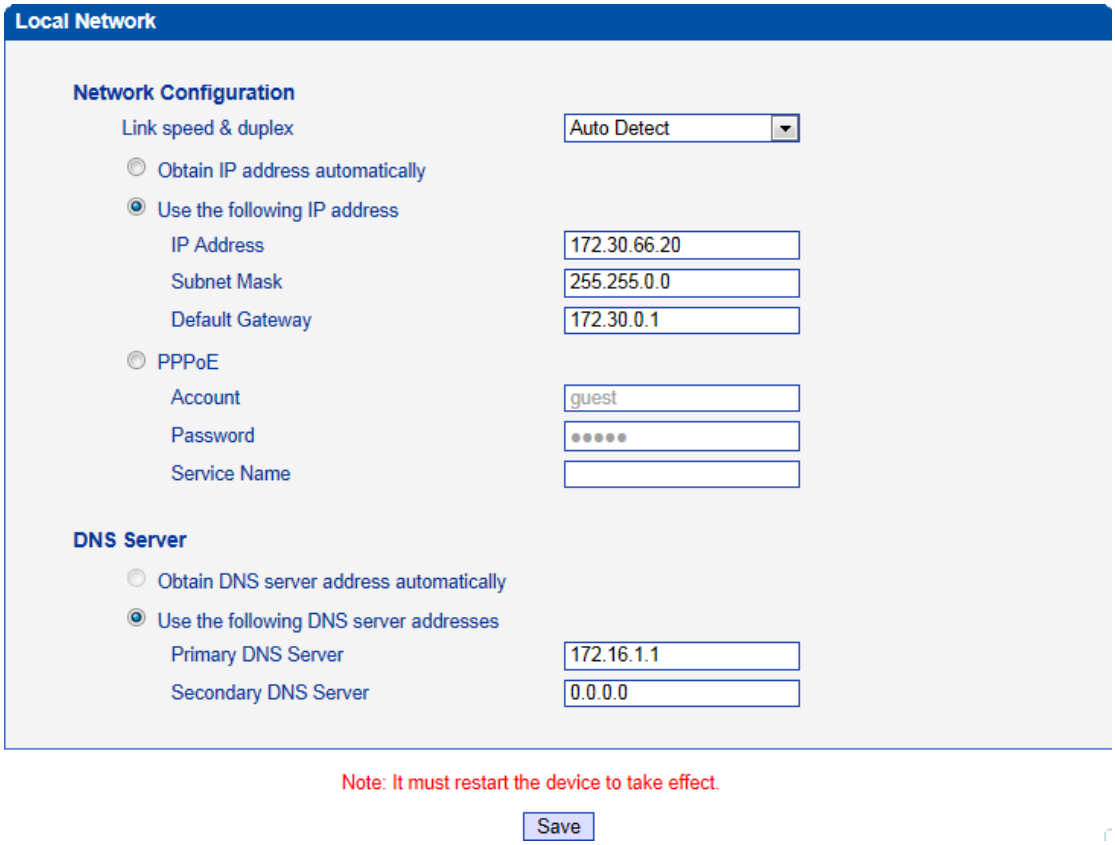

Figure 3-6-1 Local Network

Table 3-6-1 Local Network Description

Link speed & duplex	Users can select link speed and duplex in dropdown list or make the system auto detect
Obtain IP address automatically	Checked the option, the system can automatically obtain IP address by DHCP
Use the following IP address	Checked the option, users can set equipment IP address, subnet mask and default gateway.
PPPoE	Accessed Internet by this way, users need to fill in account, password and service name
Obtain DNS server address automatically	Automatically obtain DNS server IP address by DHCP
Use the following DNS server addresses	Manual input primary/ secondary DNS server IP address

Note: Please restart the device to take effect.

3.6.2 VLAN Parameter

If want to solve the problem of Ethernet broadcast and security issues, users can divide into VLAN in network. Under this configuration, the user can configure three VLAN parameters: the data VLAN, voice VLAN and management VLAN.

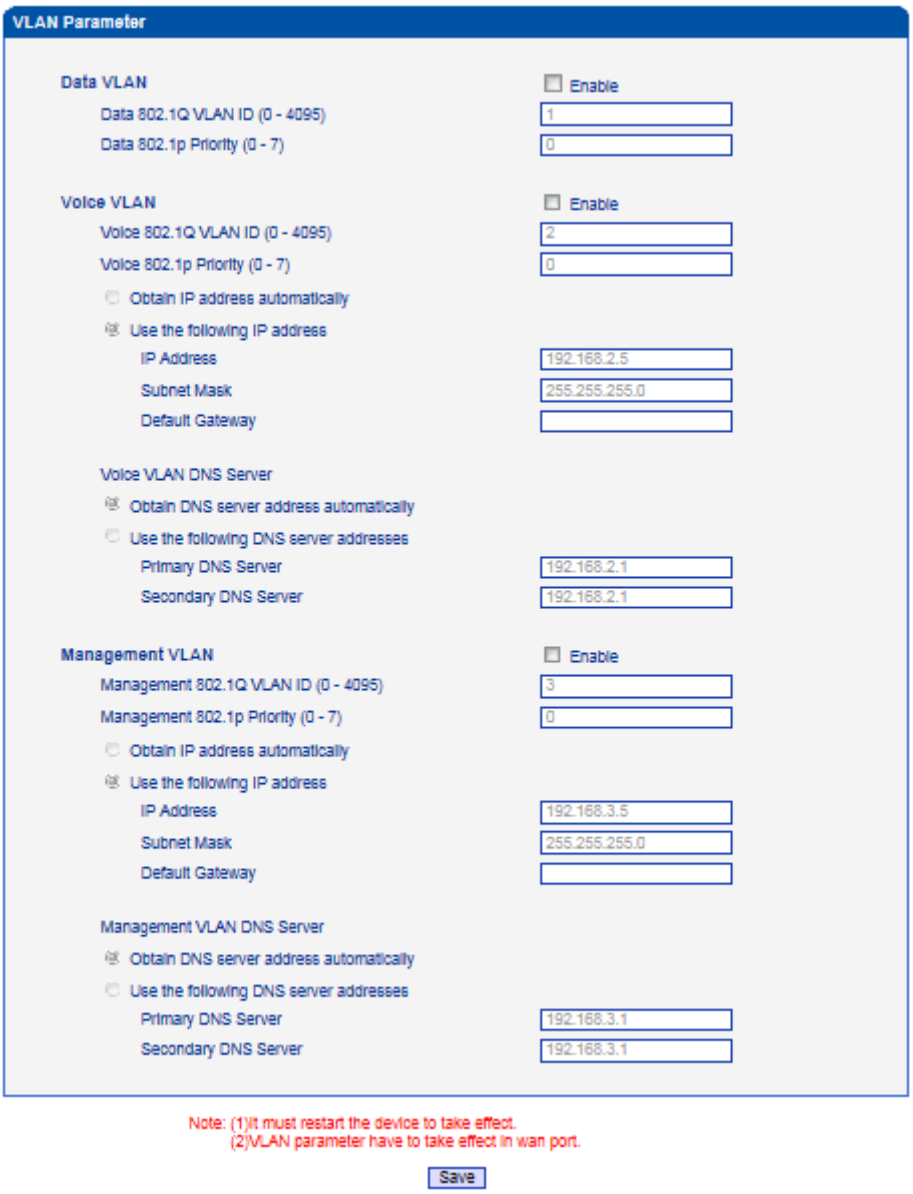

Figure 3-6-2 VLAN Parameter

Table 3-6-2 VLAN Parameter Description

Data VLAN	Data 802.1Q VLAN ID	Under standard VLAN protocol set VLAN ID. “0” is used to management VLAN, and can’t be used to service configure.
	Data 802.1p Priority (0-7)	Under 802.1q protocol users can set VLAN priority
Voice VLAN	Voice 802.1Q VLAN ID	Under standard VLAN protocol set VLAN ID
	Voice 802.1p Priority (0-7)	Under 802.1q protocol users can set VLAN priority
	IP address	Users can set DHCP or static IP address
	Voice VLAN DNS Server	Users can set DHCP or static DNS server IP address
Management VLAN	Management 802.1Q VLAN ID	Under standard VLAN protocol set VLAN ID. “0” is used to management VLAN, and can’t be used to service configure.
	Management 802.1p Priority (0-7)	Under 802.1q protocol users can set VLAN priority
	IP address	Users can set DHCP or static IP address
	Management VLAN DNS Server	Users can set DHCP or static DNS server IP address

Note: After configuration, please restart equipment to take effect; VLAN parameters must be configured in WAN port.

3.6.3 ARP

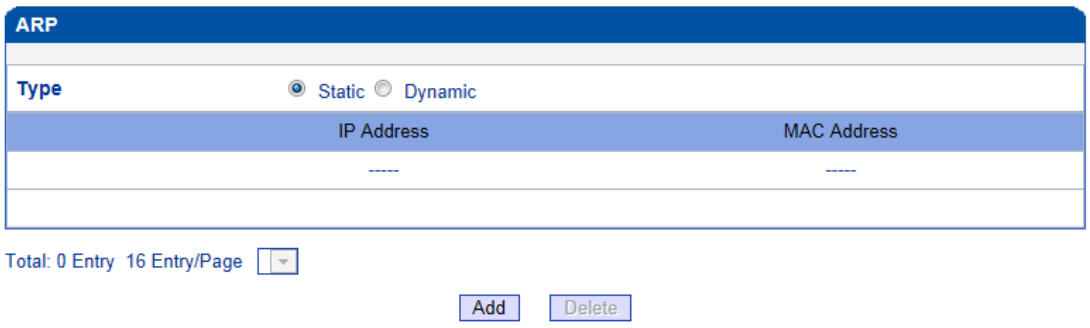

Figure 3-6-4 ARP

Customers can configure IP address and MAC address by static or dynamic.

3.7 System Configuration

3.7.1 Service Parameter

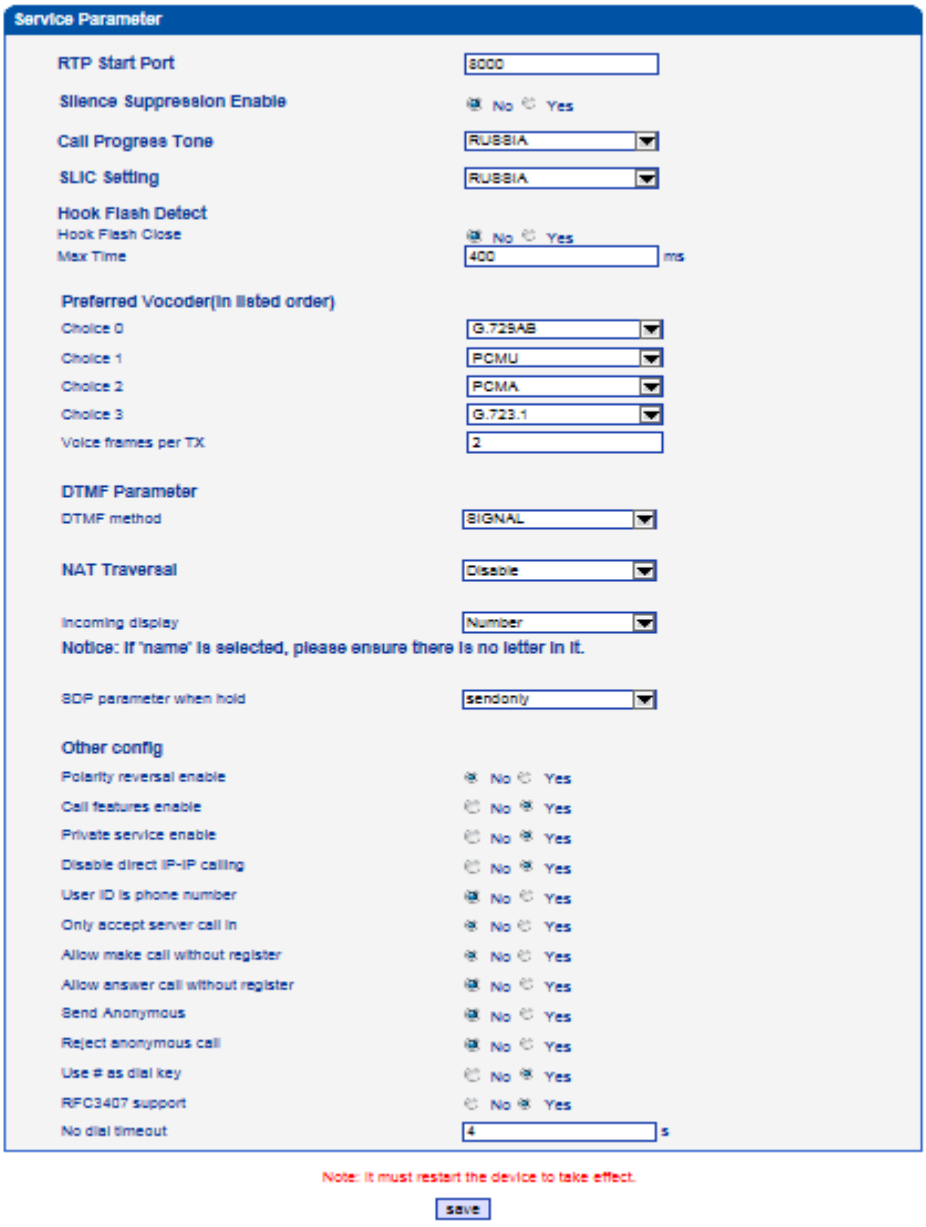

Figure 3-7-1 Service Parameter

Table 3-7-1 Description of Service Parameter

RTP Start Port	Local RTP port , default is 8000
Silence Suppression Enable	Not enable by default
Call Progress Tone	Supported many countries signal
SLIC Setting	Supported many countries SLIC standard
Hook Flash Close	Enable hook flash by default
Max Time	Max time of hook flash detect
Voice Codec	Supported G.723.1、G.729AB and PCMA/U. Customers can set codec mode according to priority.
DTMF Parameter	Supported 3 DTMF methods: INBAND, RFC2833, SipInfo
NAT Traversal	NAT function can convert equipment private IP into public IP, and help users alleviate the shortage of IP addresses through modify TCP/UDP package header IP information. The function is closed by default. NAT has two ways: STUN and static NAT. When using STUN mode, the NAT equipment doesn't need to change.
Incoming display	Users can choose to display the number or user name. If customer select the user name, please make sure there are no letters.
SDP Parameter when hold	Selected the way of playing voice stream. When hold, playing voice stream from the end, please select "sendonly". If selected "inactive", voice stream will be played from local.
Polarity reversal enable	This function is used to billing. Enabled by default.
Call features enable	If customers configure telephone function by dialing "*", then this service will be enabled. Enabled by default.
Private service enable	If customers configure call function by dialing "*", then this service will be enabled. Enabled by default.
Disable direct IP-IP calling	After enable this function, users can't call by directly dialing IP . Enabled by default.
User ID is phone number	This is the docking parameters for SIP. If selected "Yes", then user ID is just configured to phone number.
Only accept server call in	Rejected incoming call from outside the SIP server. Disenabled by default.
Allow make call without register	Enabled outcoming without register
Allow answer call without register	Enabled incoming without register
Send anonymous	With this feature enabled, caller information is not displayed when outcoming
Reject anonymous call	Rejected incoming call without caller information
Use # as dial key	After dialing the number with "#" end. Enabled by default.
RFC3407 support	Is chosen to support the SDP, the default does not support
No dial timeout	Time of no dial timeout is 4s by default

Note: After configuration, please restart equipment to take effect.

3.7.2 SIP Parameter

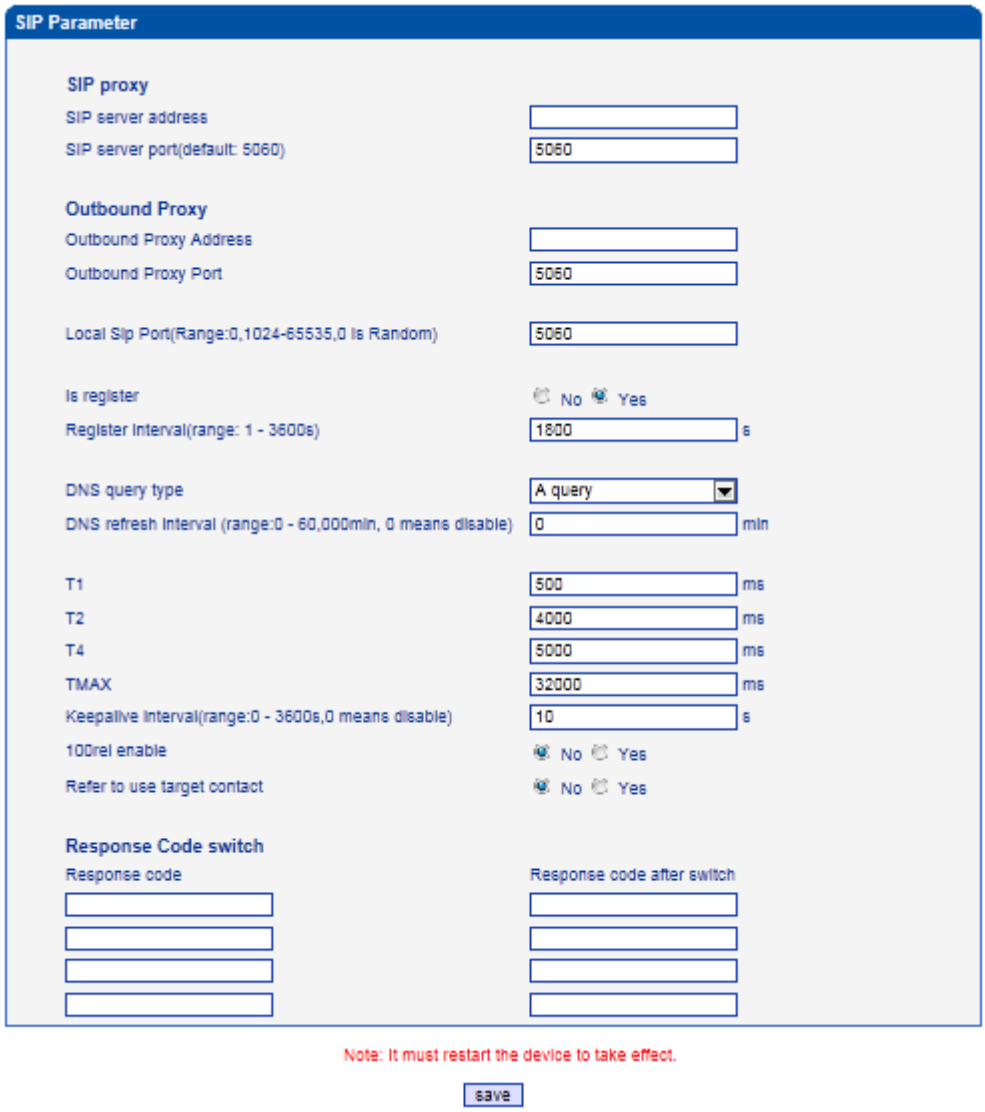

Figure 3-7-2 SIP Parameter

Table 3-7-2 Description SIP Parameter

SIP server address	SIP server IP address where equipment registered
SIP server port	Default is 5060
Outbound Proxy Address	If there is no SIP server, outbound proxy is still able to ensure the SIP message is sent to the peer. Please fill in a outbound proxy IP address
Outbound Proxy Port	Default is 5060
Local Sip Port	Local SIP listening port, users can choose a random or fixed port. 0 is random port, fixed port is 5060 by default.
Is register	Usually registration method is used. If users want to configure call without registration, please select “NO”, and allow call without registration
Register interval	The time interval of registering to SIP server or outbound proxy

DNS query type	There are two query type: A query and SRV query. When used A query, a domain name is able to return multiple IP addresses, but using only one IP address. When used SRV query, in addition to domain query information, there are TCP/UDP, port and other information. According to query information, system automatically chosen a SIP server in a group SIP server, and then obtain server IP through A query again.
DNS refresh interval	DNS information refresh time interval, 0 means no refresh.
T1	SIP protocol T1 timer value, default is 500ms.
T2	SIP protocol T2 timer value, default is 4000ms.
T4	SIP protocol T4 timer value, default is 5000ms.
TMAX	After sending SIP request, If there is no any respond in maximum time of wait for response retransmission, SIP request will be resent
Keepalive interval	Interval of sending heartbeat, it is used to communication between equipment and SIP server, and guarantee the register status of equipment. Usually use the default value.
100rel enable	SIP compatibility configuration, enabled provisional response reliably retransmission by default. When the peer sent 100rel, response PRACK to peer.
Refer to use target contact	SIP compatibility configuration, fill “contact header” value in “Refer-to” of SIP message
Response code switch	Change SIP error code according to PBX

Note: After configuration, please restart equipment to take effect.

3.7.3 Port Parameter

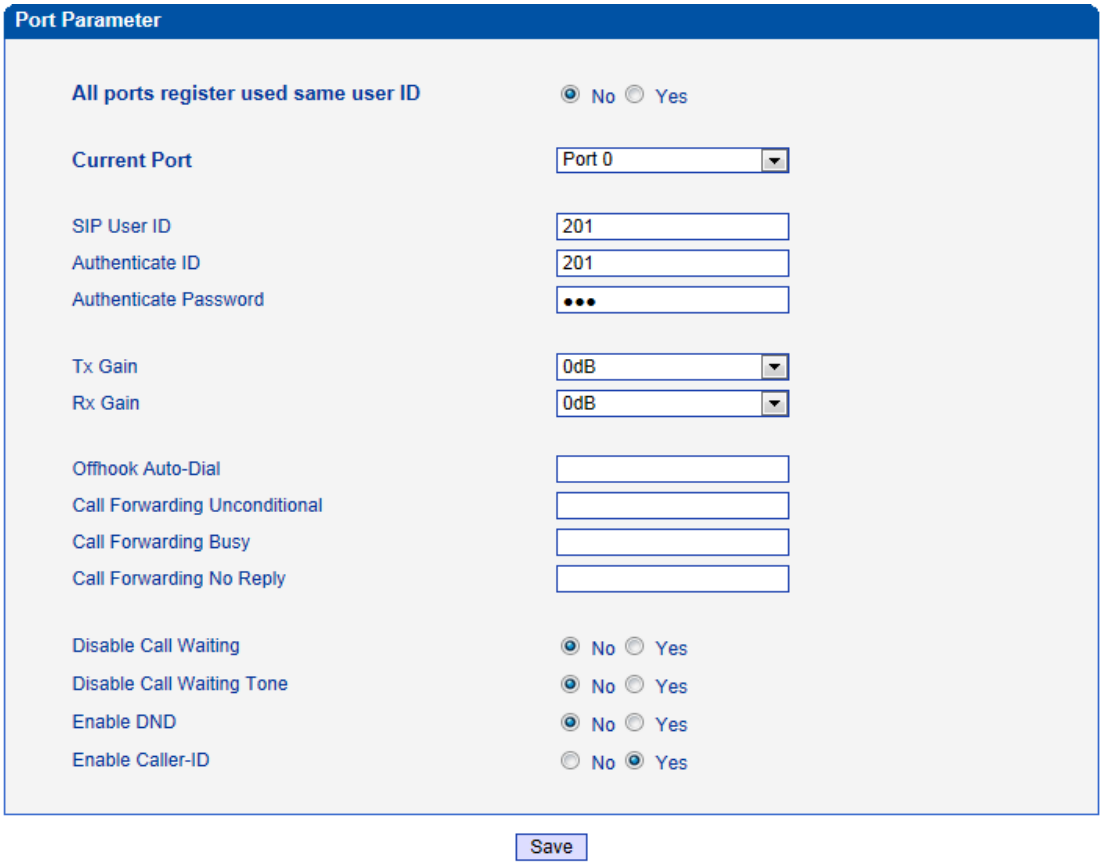

Figure 3-7-3 Port Parameter

Table 3-7-3 Description Port Parameter

All ports register used same user ID	If select “Yes”, all ports registered to SIP server by a same account; if select “No”, a port corresponds to a SIP account to register.
Current Port	If all ports register used same user ID, this option will not be configured, else current port will be port that registered to SIP.
SIP User ID	SIP user ID is user information, and displayed on the SIP server or LCD of telephone. Usually, SIP user ID is a phone number, external number or name.
Authenticate ID	Authenticate ID is the purpose of authenticate, and is used to authenticate users. SIP user ID can be the same with authenticate ID, and can also be different.
Authenticate Password	Together with the authenticate ID used to authenticate.
Tx Gain	Gain to the PSTN side outcoming
Rx Gain	Gain to the IP side incoming
Offhook Auto-Dial	Fill in a phone number. Automatically dial this number when offhook
Call Forwarding Unconditional	Fill in a forwarding number. Unconditionally transfer the call to this number.
Call Forwarding Busy	Fill in a forwarding number. Transfer the call to this number when busy.

Call Forwarding No Reply	Fill in a forwarding number. Transfer the call to this number when no reply
Disable Call Waiting	Enabled by default
Disable Call Waiting Tone	Enabled by default
Enable DND	Disabled by default, user can received all incoming call.
Enable Caller-ID	When the incoming call, display the phone number or user name, enabled by default.

3.7.4 Fax Parameter

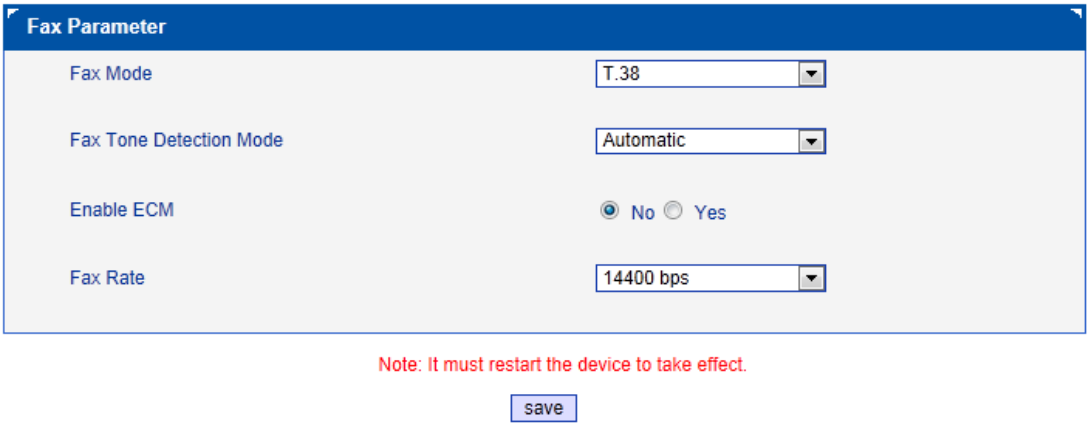

Figure 3-7-4 Fax Parameter

Table 3-7-4 Description Fax Parameter

Fax Mode	Supported two fax mode: T.38 and Pass-through. T.38 is the main protocol network fax. If equipment doesn't support T.38, please select pass-through.
Fax Tone Detection Mode	There are three modes: caller, callee and automatic. If faxes connected on the same device, in order to prevent mutual interference between devices, generally don't select the automatic mode.
Enable ECM	Enabled Electronic Counter Measures. If send unsuccessfully, device will automatic retransmission, however, it will affect the fax speed. Disabled by default.
Fax Rate	Max fax rate is 14400bps, if the baud rate is set too small to affect the speed of sending and receiving faxes and quality.

3.8 Digit Map

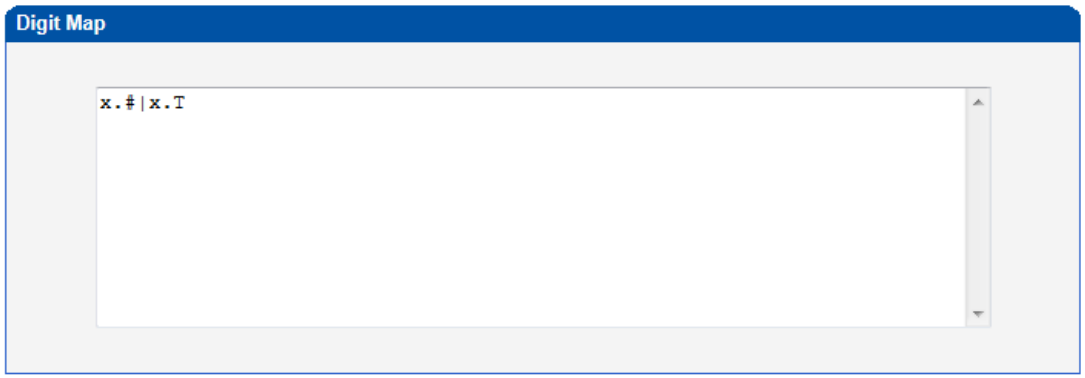
Digit Map
x.#|x.T
NOTE: Length of 'Digit Map' should be not more than 120 characters.
Save
Digit Map Syntax:
1. Supported Objects
Digit: A digit from "0" to "9".
Timer: The symbol "T" matching a timer expiry.
DTMF: A digit, a timer, or one of the symbols "A", "B", "C", "D", "#", or "*".
2. Range []
One or more DTMF symbols enclosed between square brackets ("[" and "]"), but only one can be selected.
3. Range ()
One or more expressions enclosed between round brackets ("(" and ")"), but only one can be selected.
4. Separator
|: Separated expressions or DTMF symbols.
5. Subrange
Two digits separated by hyphen ("-") which matches any digit between and including the two. The subrange construct can only be used inside a range construct, i.e., between "[" and "]".
6. Wildcard
x: matches any digit ("0" to "9").
7. Modifiers
.: Match 0 or more times.
8. Modifiers
+: Match 1 or more times.
9. Modifiers
?: Match 0 or 1 times.
Example:
Assume we have the following digit maps:
1. xxxxxxx | x11
And a current dial string of "41". Given the input "1" the current dial string becomes "411". We have a partial match with "xxxxxxx", but a complete match with "x11", and hence we send "411" to the Call Agent.
For example,the right num is 511,and the wrong num is 1111.
2. [2-8] xxxxxx | 13xxxxxxxxx
Means that first is "2","3","4","5","6","7" or "8", followed by 6 digits; or first is 13, followed by 9 digits.
For example,the right num is 2111111,and the wrong num is 1111111.
3. (13 | 15 | 18)xxxxxxxxx
Means that first is "13","15" or "18", followed by 8 digits.
For example,the right num is 13111111111,and the wrong num is 12111111111.

Figure 3-8-1 Digit Map

Under the configuration page, users can set up dialing rules that users want to dial according to dial table grammar rules. Default dial rules mean that "#" end of the dial and turn the timer.

3.9 Tools

3.9.1 Management Parameter

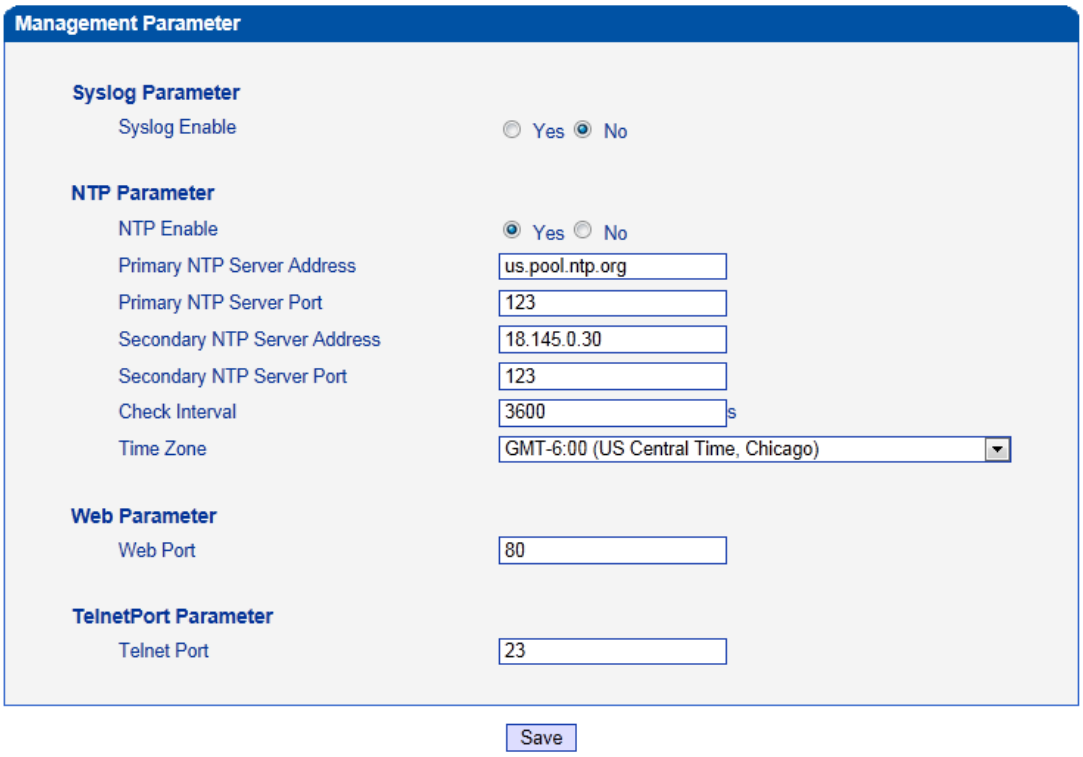

Figure 3-9-1 Management Parameter

Table 3-9-1 Description Management Parameter

Syslog Enable	Whether to enable system registry log
NTP Parameter	Enabled NTP by default, users can set addresses and ports of primary\secondary NTP server; default check internal is 3600s; users can select time zone according to the actual situation.
Web Parameter	Default web port is 80
Telnet Parameter	Default telnet port is 23

3.9.2 SNMP Parameter

Simple Network Management Protocol (SNMP) is application layer protocol, and used to manage communication line. This equipment supported three versions: V1, V2C and V3. In addition to V3 version, the other two versions do not support encryption. However, the service is usually located on the edge of the network devices, security risk, it is best to disable, to be used again.

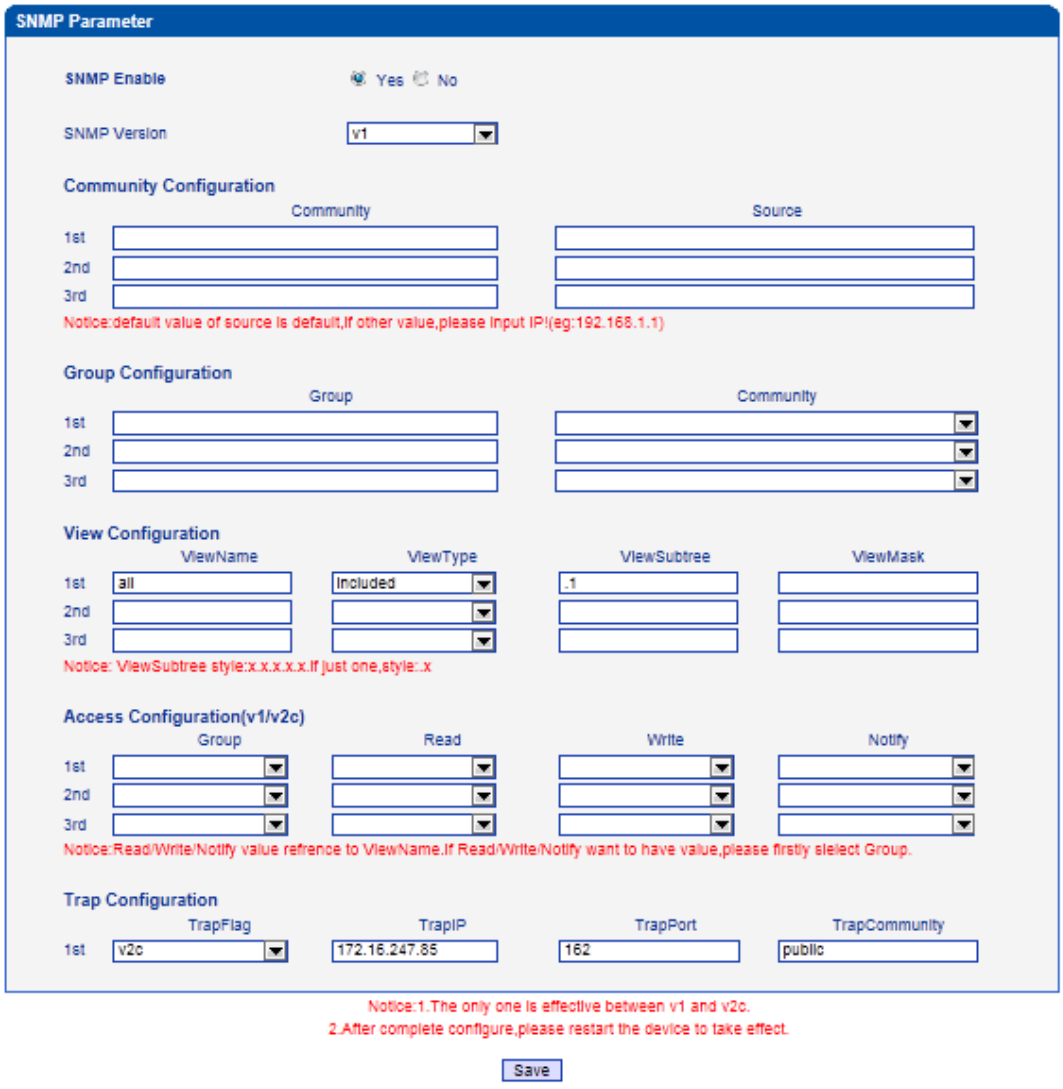

Figure 3-9-2(1) SNMP Parameter V1\V2c

Table 3-9-2(1) Description SNMP Parameter V1/V2C

Community Configuration	Community	The name of network management server managed equipment
	Source	Network management server address
Group Configuration	Group	Name of community group, different versions can use a same group name
	Community	Community join the group
View Configuration	View name	The name of description mib tree
	View type	There are Included and excluded options
	View subtree	Displayed OID of access parameters
	View mask	The same with equipment subnet mask. Generally don't configure
Access Configuration(V1, V2c)	Group	Joined community groups
	Read	Read parameters of mib view
	Write	Write parameters of mib view
	Notify	Equipment send notify parameters to NM server
Trap Configuration	Trap Flag	Version of SNMP
	Trap IP	Device to inform the NM server's IP address. The IP can be configured the same with source IP in community, also be different.
	Trap Port	Default service port is 162
	Trap Community	The same with "community" in community configuration

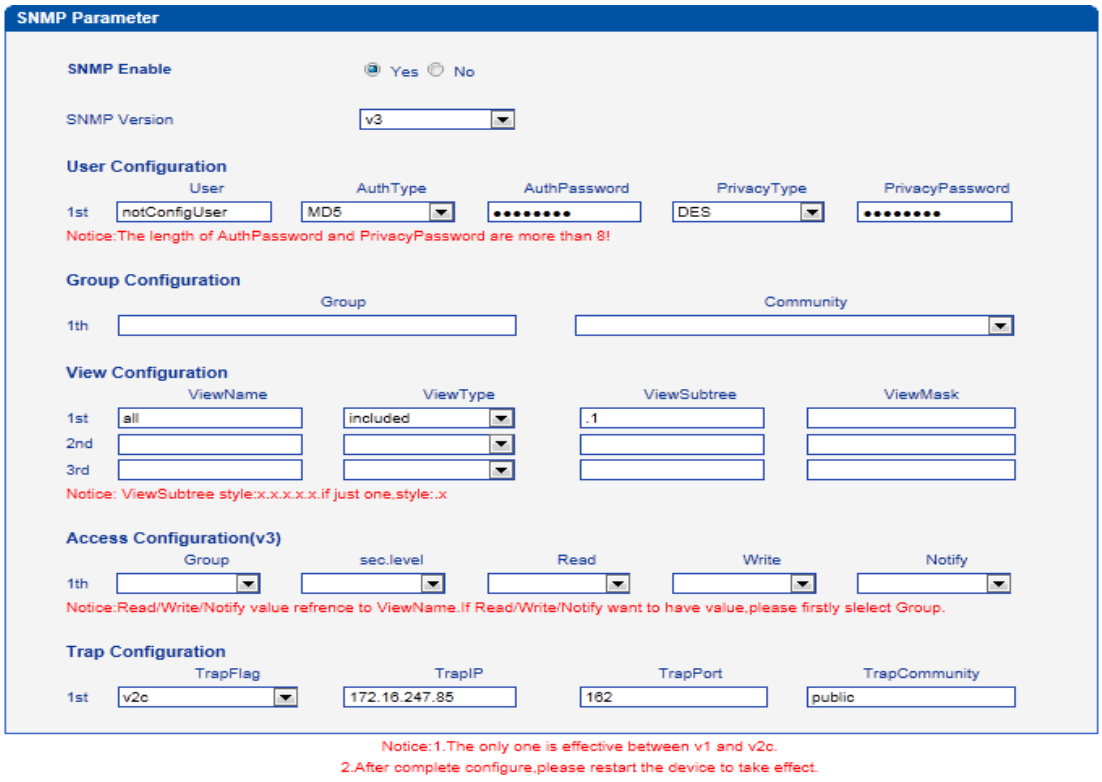

Figure 3-9-2(2) SNMP Parameter V3

Table 3-9-2(2) Description SNMP Parameter V3

User Configuration	User	Network management server management device by username
	Auth Type	Supported two auth type: MD5 and SHA
	Auth Password	Authentication password
	Privacy Type	Supported three privacy type: DES, AES and AES128
	Privacy Password	Privacy password
Group Configuration	Group	Users can use the same group name in different versions
	Community	That is user name
View Configuration	View Name	The name of description mib tree
	View Type	There are Included and excluded options
	View Subtree	Displayed OID of access parameters
	View Mask	The same with equipment subnet mask. Generally don't configure
Access Configuration	Group	Fill in group name
	Sec.level	There are two methods: authentication and authpriv. If select "authentication", users will just configure authentication information, but not privacy information
	Read	Read parameters of mib view
	Write	Write parameters of mib view
	Notify	Equipment send notify parameters to NM server
Trap Configuration	Trap Flag	Version of SNMP
	Trap IP	Device to inform the NM server's IP address. The IP can be configured the same with source IP in community, also be different.
	Trap Port	Default service port is 162
	Trap Community	The same with "community" in community configuration

Note: After configuration, please restart equipment to take effect.

Users can manage and configure gateway on remote NM server through SNMP configuration. But in order to security, recommend this option to open when needed.

3.9.3 Config Backup

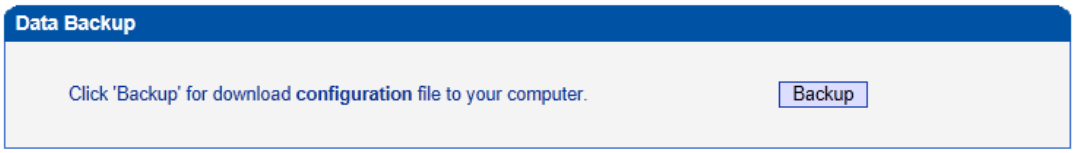

Figure 3-9-3 Config Backup

Click “Backup”, users can backup configuration information to local PC.

3.9.4 Config Restore

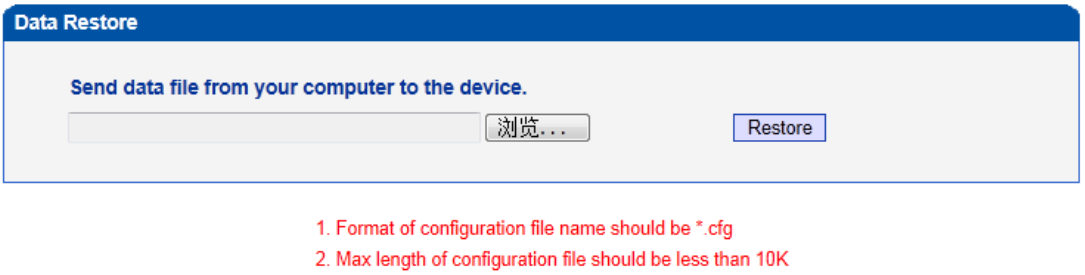
Data Restore
Send data file from your computer to the device.
浏览...
Restore
1. Format of configuration file name should be *.cfg
2. Max length of configuration file should be less than 10K

Figure 3-9-4 Config Restore

Making backup information import to device can save configuration time. After restoring, device automatically restarted.

3.9.5 Ping Test

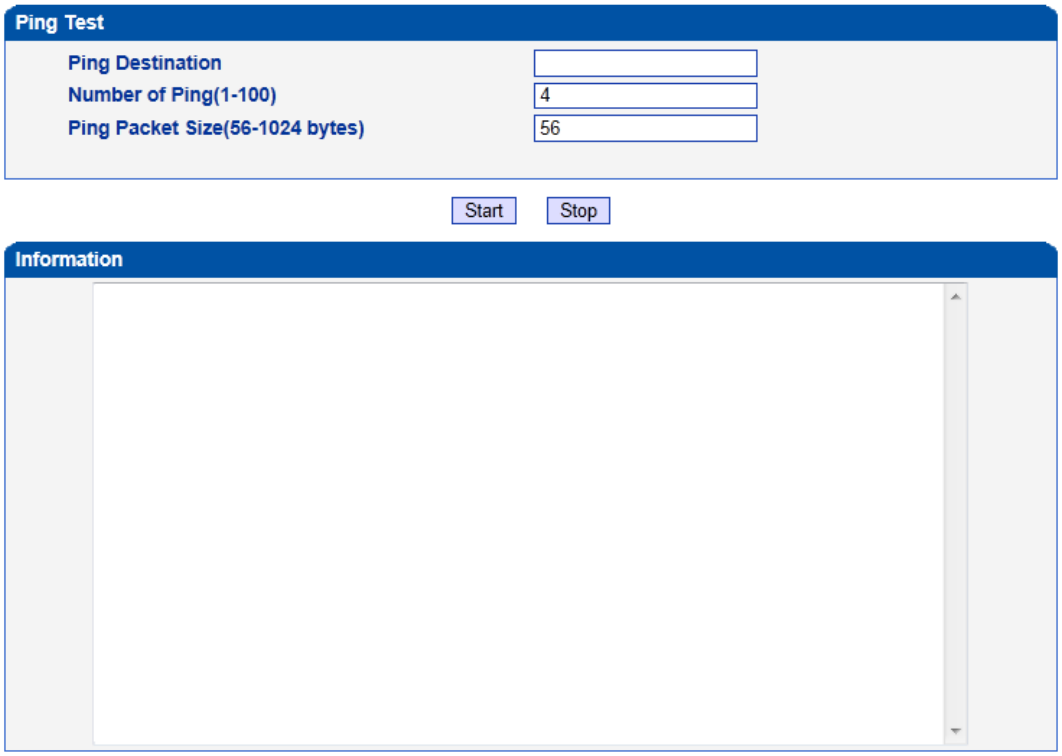

Figure 3-9-5 Ping

Filling destination IP address, numbers and length of ICMP package in this interface can test connection with destination address. Trace information is displayed in the space below.

3.9.6 Tracert Test

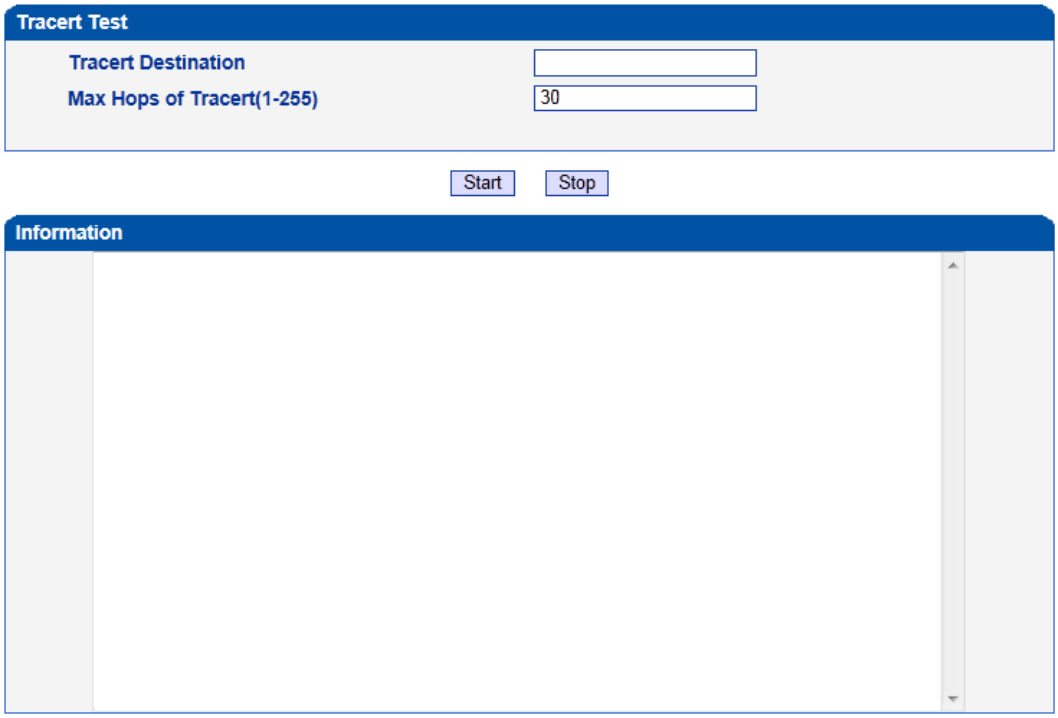

Figure 3-9-6 Tracert test

After inputting destination address and maximum number of hops, click start. Then trace information is displayed in the space below. User can see the whole route through the line and the number of hops.

3.9.7 Username & Password

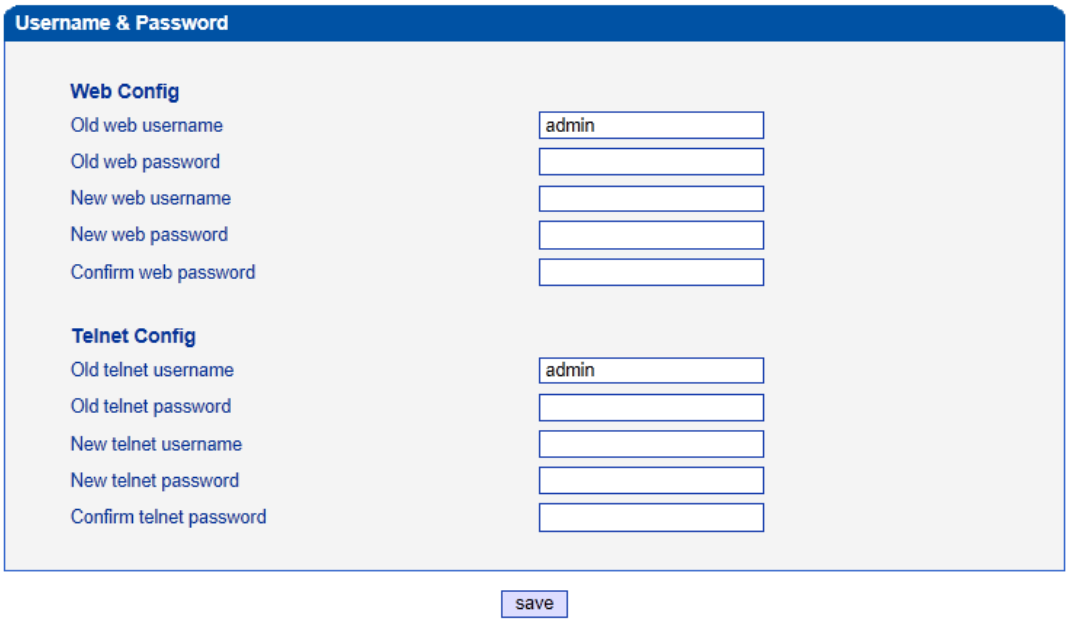

Figure 3-9-7 Username & Password

To guarantee security, users should modify web\telnet login username and password after login the web by default username and password.

3.9.8 Factory Reset

Figure 3-9-8 Factory Reset

Click “Apply” button to restore factory.

3.9.9 Firmware Upload

Figure 3-9-9 Firmware Upload

Upload “ldf” file to the device software and web upgrades. After uploading successfully, please restart device to take effect.

3.9.10 Restart

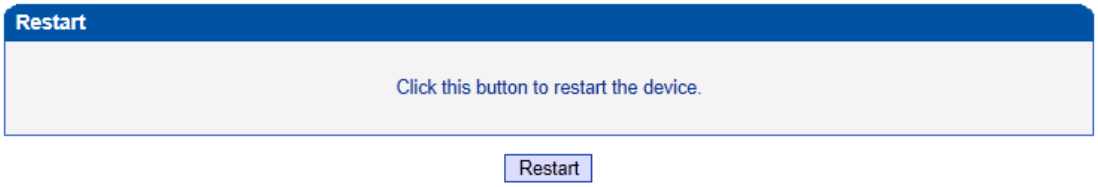

Figure 3-9-10 Restart

Click “Restart” to restart the device.

4. Term

SIP：Session Initiation Protocol

RTP: Real Time Protocol

DTMF: Dual Tone Multi Frequency

PSTN: Public Switched Telephone Network

STUN: Simple Traversal of UDP over NAT

SNMP: Simple Network Management Protocol

DHCP: Dynamic Host Configuration Protocol

VLAN: Virtual Local Area Network

NAT: Network Address Translation

SDP: Session Description Protocol

FXS: Foreign Exchange Station

FXO: Foreign Exchange Office